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MULTIPLEXING OF PACKET SPEECH ON AN EXPERIMENTAL WIDEBAND SATELLITE NETWORK

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C. J. Weinstein and H. M. Heggestad

Massachusetts Institute of Technology
Lincoln Laboratory
Lexington, Massachusetts 02173

ABSTRACT

Packet techniques provide powerful mechanisms for the sharing of communication resources among users with time-varying demands. In particular, significant potential advantages are expected for systems which integrate digital data and digitized voice in a common packet-switched network, including bandwidth savings and improvements in channel utilization and flexibility. An experimental system under development by DARPA, consisting of a wideband demand-assigned satellite network and a variety of local access networks with associated gateways, is being used to explore these expected characteristics. The components of the system are described in detail, and progress and goals in the experimental program are described.

I. INTRODUCTION

Packet techniques provide powerful mechanisms for the sharing of communication resources among users with time-varying demands. Because of their discrete and independent nature, outgoing packets can be inserted in any available time interval on a communication channel. By sharing the channel among a pool of users who each maintain a queue of packets ready for transmission, one can trade a moderate increase in overall delay for a high utilization of the channel. Packet satellite systems [1] in particular, provide the opportunity for efficient demand-assigned sharing [2,3] of long-haul communications capacity among users at widely dispersed locations. The primary application of packet techniques has been for digital data communications, where the bursty nature of user traffic can be exploited to achieve large efficiency advantages in utilization of communication resources.

Packet techniques offer significant potential advantages for voice as well. The integration of digital voice and data into a common network offers potential cost advantages [4] associated with resource sharing among different types of users, as well as the possibility for enhanced services for users who require access to both voice and data communications. Packet networks offer advantages for digital voice conferencing [5] in terms of channel utilization and control flexibility. Channel capacity savings for packet voice are achieved by transmitting packets only when speakers are actually talking (i.e., during "talkspurts"). Then, by multiplexing many packet speech users on a wideband channel a factor approaching two in bandwidth saving can be achieved, similar to that achieved on specific long-haul circuit-switched network links through Time-Assigned Speech Interpolation [6] (TASI) or Digital Speech Interpolation [7] (DSI). Other benefits are obtained with packet voice. A packet network allows convenient accommodation of voice terminals with different data rates and data formats. If vocoders are used to reduce bandwidth utilization, then each vocoder will use only the capacity necessary to transmit its information rather than the fixed minimum bandwidth increment typically used in circuit-switched networks. Packet networks also provide a convenient system environment for implementation of voice flow control techniques [8] where the bit rate is dynamically adapted to network conditions.

An experimental wideband satellite-based packet system [9] is being implemented to develop and demonstrate techniques for achieving the advantages of integrating packet voice with data in a realistic large scale network. The research supported by this system builds upon previous experiments on the ARPANET [10,11] and on the Atlantic Packet Satellite Network (SATNET) [12]. In those experiments the basic feasibility of packet speech was demonstrated, techniques were developed [10,13] for reconstitution of speech from packets arriving at non-uniform intervals, and packet protocols [14] were developed for call set-up and for speech transport. However, the network link capacities were too small to accommodate a sufficient number of simultaneous users to allow experimental demonstration of efficient statistical multiplexing of voice. The new experiment is designed around a satellite channel with a capacity of 3 Mbps, which will be able to support many simultaneous voice connections. In addition to a practical demonstration of efficient statistical multiplexing, a key purpose of the experiment is to assess the implementation requirements for terminals and for switching and multiplexing in a large-scale packet speech system.

The experimental system consists of research facilities at multiple sites which are linked by a wideband packet satellite network (the WB SATNET) including a channel on a satellite transponder, earth stations, high-performance burst modems, and demand

ignment multiple access (DAMA) processors. sites have local packet speech access ilities including packet voice terminals | (PVTs), local area networks including le [16,17] and radio [18] systems, and centrators [19,20]. The overall system ign has a high degree of flexibility, so t a variety of experiments can be supported. ides access facilities for real traffic, system includes traffic emulators to ow controlled variation of network loading. surement facilities are provided, to port experimental evaluation of system formance.

The wideband packet speech system develop-.t and the experiment program are sponsored Defense Advanced Research Projects Agency .RPA), and involve a cooperative effort by umber of organizations. These include: .t Beranek and Newman, Inc. (BBN), Cambridge, COMSAT Laboratories, Clarksburg, MD; :ormation Sciences Institute (ISI), Marina . Rey, CA; LINKABIT Corp., San Diego, CA; r Lincoln Laboratory (LL), Lexington, MA; [International, Palo Alto, CA; and Western ion, Inc., Upper Saddle River, NJ. The :ense Communications Agency (DCA) has ›ported the development of the satellite twork along with DARPA, and is utilizing : WB SATNET for a set of experiments ›porting the development of the future fense Communications System. One of the ır presently-installed network nodes is :ated at the Defense Communications Engineering nter (DCEC) in Reston, VA.

This paper describes the experimental deband packet system and the ongoing experi-nts in packet speech multiplexing. Section presents a system overview, and describes w the WB SATNET fits into a context of an terconnection (or internetwork) of local d long-haul packet nets. Specific subsystems the WB SATNET, and design considerations derlying their development, will be described Section III. Priority-Oriented Demand signment (PODA) [1], the demand assignment ltiple access (DAMA) algorithm used for ficient sharing of the satellite channel, reviewed in Section IV. Section V describes e subsystems implemented for multiplexing local area traffic, and Section VI discusses e packet voice protocols used for concentra-on of speech traffic. Section VII reports current and planned experimental activities, d a summary and conclusions are given in ction VIII.

II. SYSTEM OVERVIEW

Currently there are four WB SATNET tes, located at DCEC, ISI, LL, and SRI. At ch site the WB SATNET equipment includes an rth station, burst modem, and packet satellite MA processor. These elements are included the region labeled WB SATNET in Fig. 1, d will be described in Section III. The ırst modem and DAMA processor are designed th a high degree of flexibility so that a riety of experiments can be supported. The ; SATNET performs the task of accepting ckets from external sources, managing the tellite channel allocation on a demand assigned sis, and delivering the packets to their destination. As indicated in Fig. 1, the WB SATNET provides long-haul broadcast connectivity among several local area networks. At LL and at ISI there are local broadcast cable networks (referred to as LEXNET for Lincoln Experimental Network) which were developed at LL to efficiently support local packet voice and data traffic. LEXNET uses a carrier-sense multiple access protocol with collision detection (CSMA-CD) similar to that used in ETHERNET [21], but includes a specialized distributed access control mechanism [16], designed to take advantage of the fact that speech traffic contains bursts of periodically-generated packets. At SRI the "local" net is a packet radio network [22] which includes provision for mobile users as well as users at fixed positions.

Access from the local area onto the WB SATNET is obtained by means of an internetwork gateway at each site, as shown in Fig. 1. The gateways carry out a number of tasks including (1) communication with the WB SATNET nodes; (2) inserting the specific WB SATNET headers needed to transport long-distance packets to the destination gateway; (3) concentrating speech packets from a number of local terminals into aggregated packets for the satellite net; and (4) re-questing satellite capacity allocation on the WB SATNET based on rate requirements identified by packet voice terminals at dial-up. The channel allocation requirement is ideally set on a statistical basis and takes account of the fact that voice packets are transmitted only during talkspurts. This resource allocation function for gateways is part of the "stream" or ST protocol [23,19] which is implemented experimentally in the wideband system, and is compatible with the DoD standard internet data packet (IP) protocol (see [24]). IP gateways treat packets as independent datagrams; ST gateways provide real-time stream allocations as well as packet aggregation.

The internetwork configuration shown in Fig. 1 also includes gateway connections to the ARPANET, a long-haul store-and-forward packet-switching network connecting research facilities across the U.S. and in Europe. This connection allows intercommunication among computers (referred to as "hosts,") on the ARPANET and terminals and hosts in the other local networks. For example, the network control center (NCC) for the WB SATNET resides in an ARPANET host computer at BBN. Direct ARPANET access is provided from this BBN host computer to the DAMA processors in the WB SATNET, supporting remote monitor-ing and control functions which are necessary during installation and checkout as well as for some categories of experiments. Voice communication can be carried out between experimental packet voice facilities developed for the ARPANET and new facilities on LEXNETs and PRNETs. On a limited basis, terrestrial routing via the ARPANET (e.g., from LEXNET 1 to LEXNET 2) can be an alternative to satellite routing, provided that sufficient voice bit rate compression is applied so that

he limited throughput capability of the RPANET can handle the real-time transmission ate.

Two kinds of interfaces are shown in ig. 1 between the packet-switched system and ircuit-switched systems of the type more ommonly used for voice communication. Special circuit/packet interfaces have been developed [25] for the gateways at LL and DCEC to allow communication with a digital circuit switch in the T1 digital carrier format used for interswitch communication in digital telephony. This allows communication between voice terminals on circuit-switched and packet-switched networks, but more importantly provides the capability for experiments in which a DAMA satellite network is used as an overlay to a terrestrial circuit-switched net. These satellite overlay experiments are being implemented, under DCA sponsorship, to develop networking techniques applicable to the next generation CONUS AUTOVON [26] and a future Defense Switched Network (DSN) [27] which will utilize a mix of transmission media to provide survivable and economic digital voice and data service to DoD subscribers.

The switched telephone network (STN) interface developed at ISI allows connection between individual telephone lines (rather than switches) and packet voice terminals (PVTs) on a dial-up basis. This interface allows a user to dial into the packet voice system from any remote telephone. Plans also call for STN interfaces to be installed at other sites in the system.

With this system overview as background, the next section will proceed with a more detailed description of the functions and subsystems of the wideband packet satellite network.

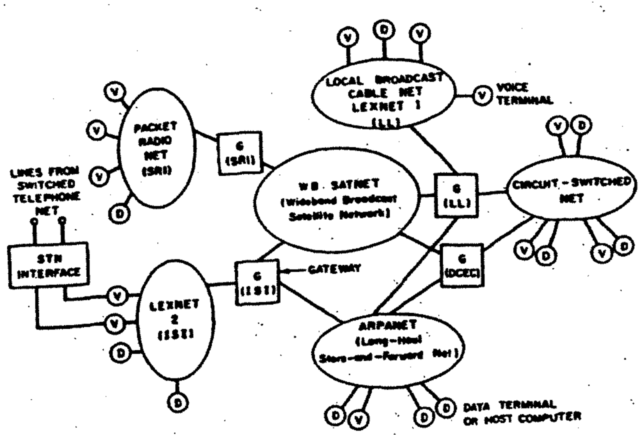


Fig. 1. Integrated voice/data packet internetwork.

III. PACKET SATELLITE NETWORK SUBSYSTEMS

The WB SATNET provides packet-switched broadcast satellite connectivity among the local networks at the four experimental sites (DCEC, ISI, LL and SRI). The major WB SATNET subsystems required at each site, illustrated in Fig. 2, are the antenna and earth station; the Earth Station Interface (ESI); and the Pluribus Satellite Interface Message Processor (PSAT). The Lincoln Laboratory site configuration, including the WB SATNET subsystems as well as a concentrator and a local network, is typical of the wideband experiment sites; it is illustrated in Fig. 3.

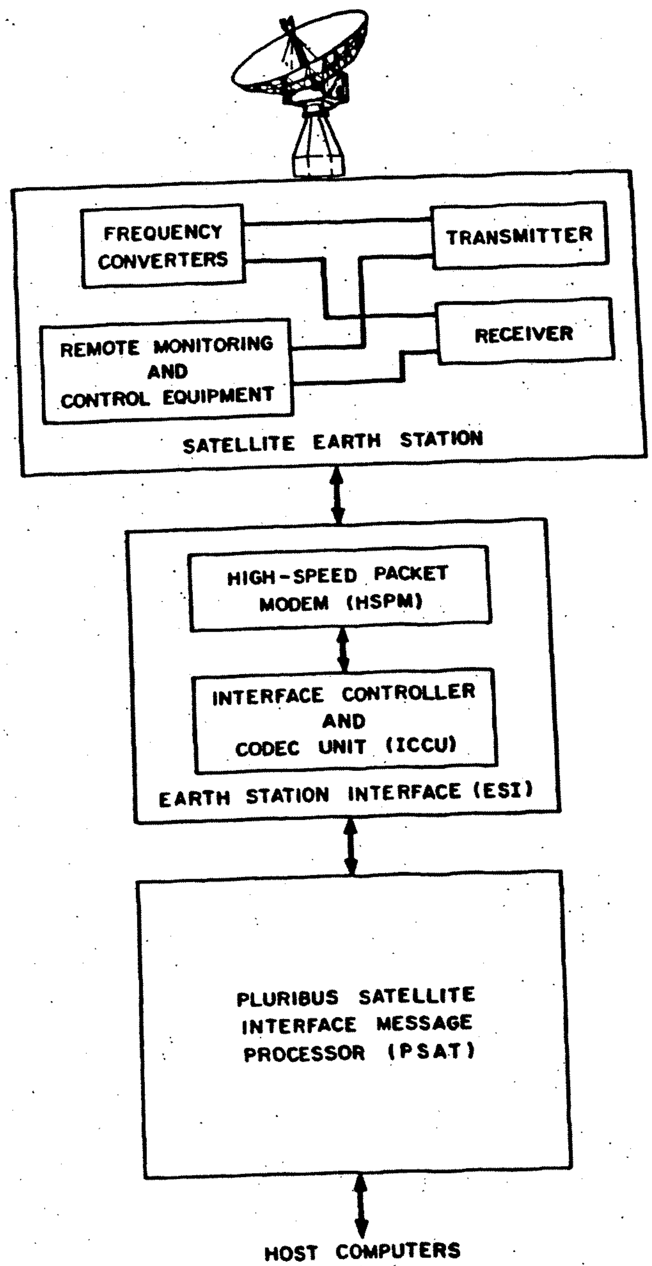


Fig. 2. Block diagram of wideband packet satellite network subsystems.

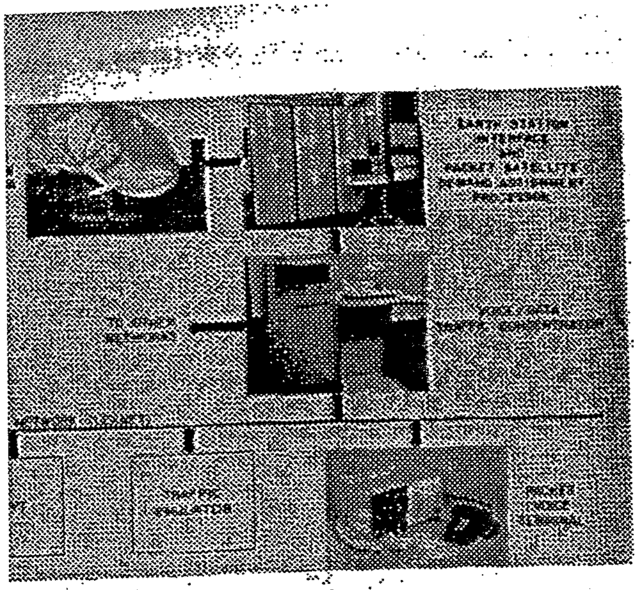


Fig. 3. Typical site configuration for experimental wideband packet network.

The satellite link, antenna and earth ation subsystem are leased from Western ion, Inc. Satellite capacity is provided the 4/6 GHz commercial telecommunications ad. The WB SATNET carrier is at the upper l of Transponder 1 on Western Union's STAR III satellite, which is located at out 91 degrees west longitude. The exact equencies are 5.959 GHz (uplink) and 3.734 z (downlink), and the available bandwidth matched to the 3.088 Mbps digital signal ed by the system. The antenna (a 5-meter ssegrainian) has a field-effect transistor ET) receiver front-end amplifier mounted ind its feed horn; together they provide earth station figure of merit G/T of about dB/Kelvin. Uplink and downlink power dgets are designed to support system operation a specified maximum bit error rate of x 10^{-7} at a channel bit rate of 3.088 Mbps. e transmitter amplifier is a 125-watt aveling-wave tube (TWT); it is installed in small shelter at ground level behind the tenna in Fig. 3, along with suitable frequency - and down-converters and IF cable drivers d receivers. In order for the equipment to operated as an unmanned earth station, it provided with remote monitoring and control cilities connected via automatic telephone aling equipment with a central satellite ntrol facility in New Jersey.

The earth station interfaces at 84 MHz a analog IF cables with a flexible burst dem (the High-Speed Packet Modem or HSPM). is subsystem together with the Interface ntroller and Codec Unit (ICCU) constitutes e Earth Station Interface (ESI) equipment, ich is manufactured by LINKABIT. The HSPM fers a choice of binary or quaternary ase-shift keying (BPSK or QPSK), at channel mbol rates of 193, 386, 772, or 1,544 lobaud. The bit rate with QPSK is twice the channel symbol rate, and hence the maximum bit rate over the channel is 3.088 Mbps. The Codec implements convolutional encoding and sequential decoding at a selection of code rates (1, 7/8, 3/4 and 1/2) with corresponding coding gains equivalent to signal-to-noise ratio improvements of 0 dB to as much as 5 dB. Suitable combinations of bit rate and code rate can be chosen to accommodate a broad range of bit error rate tolerance levels; for example, TDMA framing and synchronization information and packet headers must be received very accurately, and hence should be more heavily protected with coding (i.e., a low code rate should be used). The portions of packets which contain speech signals can tolerate increased bit error rates, on the other hand, and therefore permit operation at a higher code rate with a corresponding gain in channel throughput. Reduced code and symbol rates can also be chosen to permit interoperability with less sensitive earth stations, having smaller antennas or higher receiver noise temperatures. An important flexibility feature of the ESI is that it supports multiple changes in code rate and bit rate within a TDMA burst. Implementation of burst formats and control of the functions of the ESI are accomplished in a Motorola MC68000 16-bit microprocessor in the ICCU.

The format and contents of the transmissions in each TDMA frame are supplied to the ESI by the Pluribus Satellite Interface Message Processor, or PSAT [28]. The PSAT is a high-throughput multi-processor computer built by Bolt Beranek and Newman. The primary functions of the PSAT are management of demand-assigned allocation of time slots on the satellite channel, and transmission and reception of information bursts over the channel. The Pluribus architecture combines the processing power of six Lockheed SUE processors. Satellite demand assignment is arbitrated by a distributed protocol which combines features of both circuit- and packet-switched control techniques. This protocol, which is called the Priority-Oriented Demand Assignment (PODA) algorithm [1,28,29], is discussed in the following section.

IV. SATELLITE CHANNEL DEMAND ASSIGNMENT ALGORITHM

The Priority-Oriented Demand Assignment (PODA) algorithm was developed for the Atlantic Packet Satellite Experiment [30] (the Atlantic SATNET) which, as noted in Section I, links a number of earth stations in the United States and Europe by means of a 64 kbps INTELSAT satellite channel. The implementation of PODA for the WB SATNET, which includes a number of modifications to adapt to the higher bit rate of the channel, is described in detail in [28].

The PSAT's transmissions basically constitute a Time-Division Multiple Access (TDMA) structure with variable frame format and fixed frame duration. A dynamic channel-sharing discipline such as PODA undertakes to achieve high efficiency on the channel for

ıe relatively low-duty-cycle traffic which ; characteristic of WB SATNET users by ynamically re-allocating unused capacity nong the subset of users who have traffic ırsts to send at any given time. Traditional ixed-format TDMA and Frequency-Division ıltiple Access (FDMA) are static channel-ıaring disciplines which permanently allocate apacity among the community of users. ieally, these capacity allocations are sized o match the users' communication requirements; n practice the efficiency of utilization of he channel can be quite low, however, ecause random variations in user activity end to leave the assigned blocks of capacity mpty some fraction of the time. A user's ata stream can be communicated very efficiently sing fixed-format TDMA or FDMA if it has a igh duty cycle, corresponding to (for example) ultiplexed signals from a large local population f voice or data terminals. A user data tream containing multiplexed traffic from a oderate to small local population of terminals ill result in bursty traffic with a lower uty cycle, which is best handled by a dynamic echnique such as PODA.

Key functions that must be accommodated ıy a dynamic channel-sharing discipline nclude user capacity request mechanisms, letermination of assignments in response to he requests, and achievement of low assignment lelay coupled with efficient channel usage. dditional functions that would further nhance a dynamic discipline include the ıbility to adapt to both error-tolerant raffic (e.g., speech signals) and error-intolerant traffic (e.g., computer file ransfers), and the ability to accommodate ooth strong stations and weaker stations (e.g., those which have smaller antennas or less sensitive receivers, or are temporarily suffering reduced performance due to rainfall ıttenuation or component degradation). The nanner in which PODA incorporates these functions is described briefly below.

PODA provides two mechanisms for user capacity requests: use of a special reservation subframe, and "piggybacking" new requests ıpon ongoing transmissions. Two types of reservation subframe techniques have been implemented: fixed PODA (FPODA), in which each user is permanently allocated a specific request slot, and contention PODA (CPODA), in which request slots are shared among users on a contention basis. PODA is presently implemented as a distributed-control discipline: Each PSAT hears all capacity requests, and all the PSATs arrive at the same set of slot assignments by applying the identical algorithm to the set of requests. This results in a two-hop delay for transmissions from any given user (one satellite round trip time for submitting the capacity request, and one round trip time for actual transmission of the information in the assigned slot). Centralized assignment, in which one processor computes slot assignment for all users, results in a three-hop delay because of the extra round trip time required for the central controller to communicate its slot assignment decisions back to the requesting stations. The PODA design includes centralized assignment for a limited subset of users, namely weak stations which cannot hear all of the transmissions exchanged across the network. This centralized assignment capability is planned but not yet implemented. The PSAT can accommodate weak stations by a combination of the centralized assignment feature of PODA and the capability to make suitable code and bit rate selections by commands to the ESI.

Figure 4 is a generalized illustration of a PODA frame (typically 21.2 msec in duration in the present implementation). The information and control subframes are used as described above. Within the information subframe there are sequences of information bursts, each from a PSAT which has obtained a slot assignment in that frame. An information burst can consist of either datagram packets or concentrated stream packets. Datagrams are independent self-contained messages which are communicated between concentrators and the PSAT in accordance with the DoD standard internet data protocol (IP); they are used for delay-tolerant digital data communications such as computer-to-computer file transfers, and can offer error-free service through the use of mechanisms such as acknowledgement and retransmission. Stream packets are elements of extended real-time communication sessions such as digitized speech streams, which have the properties that (1) they will tolerate little delay, and little or no delay variation; (2) once initiated, they may be expected to continue for a time interval extending over many PODA frames. Streams are handled in accordance with a new experimental protocol called ST, which is discussed further in Section VI. PODA accommodates streams by offering stream reservations (initiated by means of requests made by the concentrator to the PSAT at call set-up time) for an extended sequence of packets providing transmission for the stated average bit rate for the duration of the call. The bit error tolerance of voice avoids the need for re-transmission, thereby tending to stabilize the delay. The gateways shown in Fig. 1 enhance the efficiency of satellite channel utilization by providing the PSAT with concentrated stream packets which have been subjected to destination-oriented aggregation, as explained in Section VI.

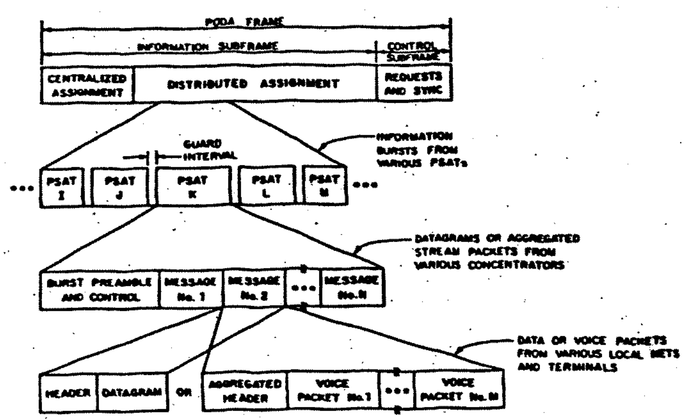


Fig. 4. Frame structure for Priority-Oriented Demand Assignment (PODA).

V. LOCAL AREA NETWORK SUBSYSTEMS

A block diagram of the local area network ›stems for a typical site is shown in 5. The concentrator/gateway has been .oped by LL with initial application as a ›T/WB SATNET traffic concentrator. It has augmented with the required network rface software and hardware to serve (in erent configurations) as a gateway to ARPANET, PRNET, and a digital circuit-ched net as well. The hardware consists central gateway processor (Digital pment Corporation PDP-11/44 minicomputer), network interface processors (UMC-Z80 ersal interface computers produced by ciated Computer Consultants) and special ware interfaces for each attached network. e the bulk of the gateway traffic is to orwarded, the data portions of packets not be examined by the gateways. The :way design also provides for an additional ›ry to be accessed by the Z80 processors a multiport basis. This shared memory is ended to store the data portions of packets that the load on the PDP-11 UNIBUS and ›ry will be minimized.

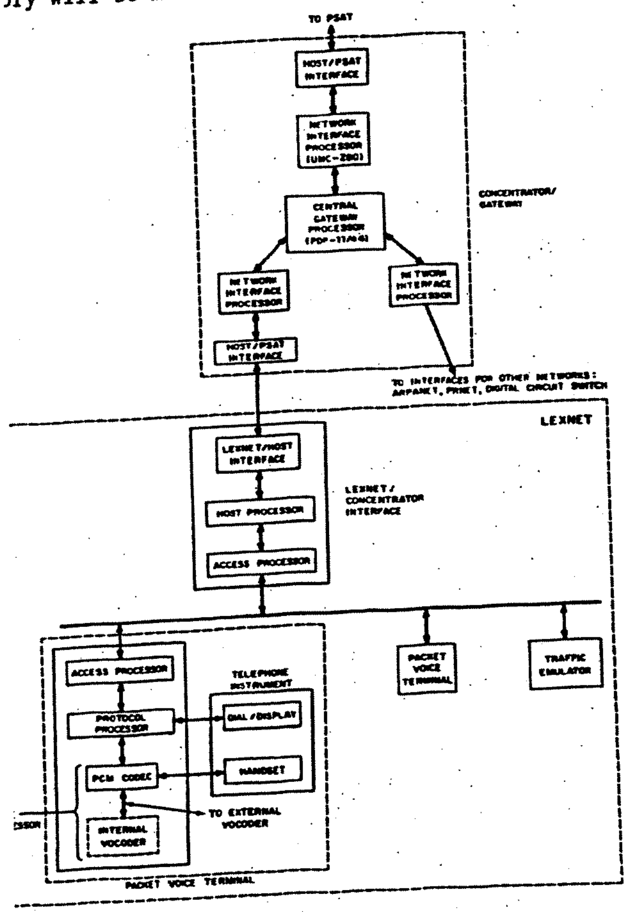


Fig. 5. Block diagram of local area network subsystems.

A functional block diagram for the ›ncentrator is shown in Fig. 6. ST gateway ›nctions deal with the processing of stream ..e., voice) traffic, including the aggregation of local area voice packets for efficient multiplexing on the WB SATNET. IP gateway functions handle datagram traffic, including the control packets needed to set up voice calls between packet voice terminals. Other functions include conference access control and control and monitoring. As indicated, the PDP-11 is the central gateway processor and the UMC-Z80s perform front-end packet handling tasks for each network. The WB SATNET UMC-Z80 performs the additional task of deaggregating ST packets destined for different terminals on LEXNET.

The LEXNET [17] is a baseband CSMA cable network with distributed control similar to ETHERNET [21]. It utilizes a distributed algorithm for randomized retransmission specialized for voice traffic [16]. This algorithm estimates competing network activity based on the fact that a voice terminal produces periodic packets during talkspurts, in contrast to data terminals where the inter-packet intervals are usually modelled as being independently distributed. The range of the retransmission interval after collisions is adjusted along with this activity estimate, with longer intervals corresponding to higher activity estimates.

The Packet Voice Terminal (PVT) [15] has been developed with a flexible and modular architecture, as illustrated in Fig. 5. There are three primary functional units, each controlled by a microprocessor. The voice processor digitizes the speech but is independent of the transmission medium. Each PVT has a 64 Kbps PCM capability and a selectable option of internally mounted vocoders of various rates, or connection to an external vocoder. The protocol processor forms the voice packets and provides the necessary layers of Network Voice Protocol (NVP) [14,19] to insure that the packet can be delivered to a distant network and played out at the proper time. The necessary buffering and speech reconstitution algorithms to produce steady speech to the listener despite asynchronous packet arrivals from the network are implemented in the protocol processor. The protocol processor also includes an interface with the user dial/display and must generate and interpret the packets necessary for establishing the call. A generalized packet interface is provided between the protocol processor and the access controller, which performs LEXNET-specific local net packet transport functions. It is possible to connect the PVT to other packet networks such as the PRNET by changing the design of the access module.

In addition to the PVTs, a traffic emulator is also provided in the experimental network to allow performance measurements to be made on the network with controlled amounts of traffic. The LEXNET traffic emulator module actually consists of a PVT with special protocol processor software to generate controlled traffic.

Packets entering or leaving LEXNET pass through the LEXNET/Concentrator Interface (LCI), a unit which is very similar to the

VT. The access processor in the LCI is dentical to that in the PVT, and the LCI ost processor controls packet forwarding to nd from the concentrator.

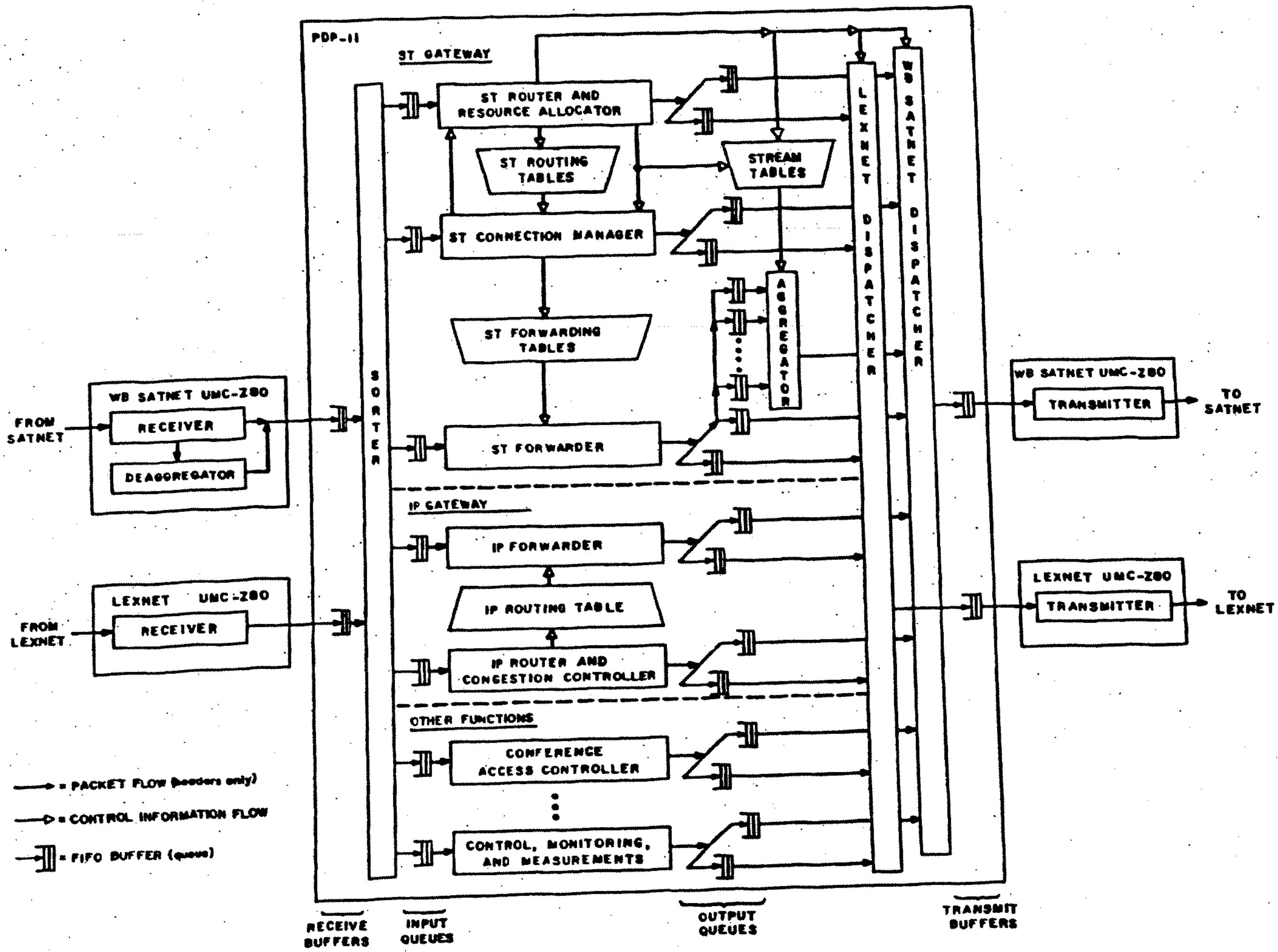
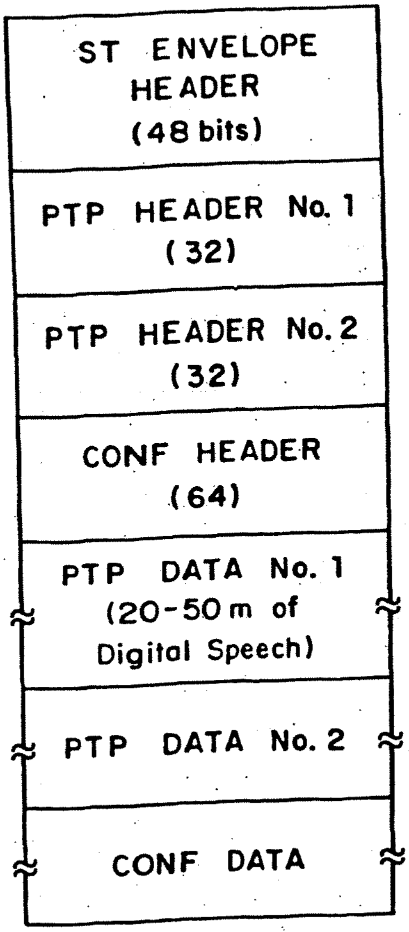


Fig. 6. Functional diagram of speech concentrator.

VI. PACKET SPEECH CONCENTRATION PROTOCOL

The format for an aggregated stream (ST) packet, as would be delivered by a concentrator to a PSAT to be transmitted in a stream slot allocated within the PODA frame, is illustrated in Fig. 7. The sample packet includes voice for two point-to-point (PTP) calls and one conference (CONF) call. All headers are grouped at the start of the packet. This allows advantage to be taken of the flexible coding properties of the burst modem (see Section III), by coding the headers (where bit errors are extremely damaging) more heavily than the speech (where some bit errors can be tolerated). The individual PTP headers are quite short, since they need contain only a connection identification (CID) and not the full destination address. The ST connection process includes the set up of tables in gateways which associate CIDs with the next forwarding address along the route. The CONF header includes additional information to control the forwarding of packets to all conference participants. The data portion of the packets include a short (32-bit) NVP header with a time stamp to allow the PVT to determine the proper packet playout time. The number of speech bits depends on the voice bit rate and the interpacket interval. Typical ranges are 2.4-64 Kbps for voice bit rate, and 20-50 msec for the interpacket interval.



g. 7. Format for aggregated stream (ST) packet.

I. CURRENT AND PLANNED EXPERIMENTAL ACTIVITY

A preliminary experiment plan [31] was epared soon after the inception of the perimental wideband packet system program. ncoln Laboratory has since been responsible r experiment planning and coordination for e overall program. Central to the experiment ans has been the notion of multiplexing fficiently large numbers of packet speech ers onto a common satellite channel to rmit testing of the multiplexing efficiency. e experiments follow a logical sequence ich has been keyed to achievement of successive ages of experimental capability, as the plementation of the various subsystems is mpleted.

The major experimental activity which s been completed at this writing has involved tegration and performance demonstration in e two primary subdivisions of the system, mely the WB SATNET and the various local cess facilities. The system is at the point beginning to support the overall goal of e program, which is to carry out a broad nge of multi-user packet speech experiments. ese will include interconnections among ice terminals within local networks; among ice terminals on similar but separate networks, internetted with each other via the WB SATNET; and among different but compatible types of voice terminals on dissimilar local networks internetted via the satellite net.

In the area of local access network integration and performance demonstration, the checkout of Packet Voice Terminals (PVTs) on LEXNETs has been substantially completed. Demonstration of two simultaneous conversations involving four PVTs on the same LEXNET was accomplished some time ago. Conversations between two PVTs on the same LEXNET have been looped back through the PSAT, ESI, and satellite transponder. A number of experiments have also been done in which PVTs on two separate LEXNETs have communicated with each other by way of a concentrator. Compatibility of the voice terminal and concentrator protocols at different sites (i.e., LL and ISI) has been established via narrowband (2400 bps) packet speech experiments using the ARPANET to link the distant local access facilities.

In the area of WB SATNET integration and checkout, the main thrust has been to achieve reliable operation of a network of PSATs performing their timing, synchronization and demand assignment functions via the satellite channel. This activity has been substantially completed with respect to a two-PSAT network (namely, those at ISI and LL).

The immediate goal at this time is to demonstrate multiple simultaneous cross-country conversations. Competing emulated traffic will then be introduced, and its intensity will be increased until system performance breakpoints can be observed. A broad range of variations on these experiments will be conducted, thus making it possible to evaluate the utility of packet switching for handling real-time multiple-user speech communication.

VIII. SUMMARY AND CONCLUSIONS

The background and motivation for experimentation with multi-user packet speech were described; cost advantages are anticipated, as well as improvements in flexibility and transmission channel utilization efficiency. The experimental wideband satellite-based network which is being implemented to support such experiments was described in some detail, including overall system design issues as well as the characteristics and design considerations related to each of its constituent subsystems. The functions and performance goals of the PODA algorithm were described in terms of DAMA implementation in the WB SATNET. Local area access facilities were described, including the LEXNET and packet voice terminals (PVTs), the Packet Radio net, and circuit-to-packet interfaces, as well as the necessary concentrators and gateways and the associated protocols. Current and planned experimental activity with the wideband system, were described including the integration and performance verification of the major subsystems as well as the near-term goals for experiments combining multiple simultaneous real and emulated conversations over the satellite.

ACKNOWLEDGEMENT

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